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Applicants McCallister, et al.

Serial No.: 10/718,505

Filed: November 20, 2003

Group Art Unit: 2611

Examiner: CORRIELUS, Jean B.

For: CONSTRAINED-ENVELOPE TRANSMITTER AND METHOD THEREFOR

INVENTOR'S DECLARATION

I, Bruce A. Cochran, a named inventor in the above-identified reissue application, hereby state and declare as follows:

I have read the "Inventor's Disclosure and Declaration Under 37 C.F.R. 1.56" of Ronald D. McCallister dated April 28, 2009

I understand and fully agree with the technical analysis and conclusions of Mr. McCallister's submission regarding the disclosure of the May et al. reference and its applicability to the present application and claims.

I am not currently affiliated with either the assignee of the present application, Mr. McCallister or Mr. McCallister's current company and I have no interest in the application.

I further declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true, and further, that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code, and that such willful false statements may jeopardize the validity of the application or any patent issuing thereon.

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Dated: April 30, 2009

Applicants	McCallister, et al.	Group Art Unit:	2611
Serial No.:	10/718,505	Examiner:	CORRIELUS, Jean B.
Filed:	November 20, 2003		
For:	CONSTRAINED-ENVELOPE TRANSMITTER AND METHOD THEREFOR		

INVENTOR'S DISCLOSURE AND DECLARATION UNDER 37 C.F.R. 1.56

I, Ronald D. McCallister, a named inventor in the above-identified reissue application, make the following disclosure and declaration pursuant to my obligation under 37 C.F.R. 1.56 to make known to the Patent Office any information believed material to the issue of patentability or that refutes or is inconsistent with a position the applicant takes in asserting an argument of patentability. I am not currently affiliated with the assignee of the application and have no interest in the application.

The assignee has repeatedly questioned my integrity in its submissions to the Examiner and the Board of Patent Appeals and Interferences, and thus I make this submission in the form of a declaration to remove any doubt that the purpose of my submission is to satisfy my Rule 56 obligation. The assignee apparently believes that pecuniary interests can trump an inventor's duty of candor to the Patent Office. I disagree vehemently with the assignee's view of the patent application process and my obligation as a participant in that process to protect its integrity. I also observe that the assignee's expert, Mr. Birch, has presumably been compensated for his services. If the assignee is so concerned with alleged "interest" by declarants, perhaps it should withdraw Mr. Birch's declarations. The same goes for Mr. Bernkopf's declaration; he is the assignee's counsel and is clearly compensated by the assignee.

Moreover, notwithstanding the assignee's repeated misstatements, my current company does not need a license under this application and had no need for a license in 2003 when the assignee asserts that my company requested such a license. The assignee implies that my current company sought a license because of infringement concerns. That is not true. Mr. Bernkopf asked whether my company would consider licensing our intellectual property at the same time we asked about potentially licensing the assignee's intellectual property. This was a general discussion including, inter alia, potential cross-licensing intended to address possible business cooperation and did not come about because either party believed it was infringing on any valid existing or future patent claims of the other party. In fact, I clearly stated in my email to Mr. Bernkopf that my company's "new approach does not infringe" the assignee's patents. Having moved well past those business discussions in 2003, my purpose in providing this submission (and my prior submissions) is merely to correct misrepresentations made by the assignee in an application on which I am a named inventor.

Initially, the Examiner's acknowledgement of my prior submissions in co-pending application Serial No. 10/718,507 dated July 5, 2005, August 16, 2006 and November 6, 2007 is appreciated. Because of subsequent positions taken by the assignee which conflict with the scope of what I and my co-inventors actually conceived and the teachings of the prior art, I believe it is necessary to make this further submission pursuant to my disclosure obligation.

My disclosure principally concerns the May et al. prior art reference ("Reducing the Peak-to-Average Power Ratio in OFDM Radio Transmission Systems," published May 18, 1998 in the Proceedings of the 1998 Vehicular Technology Conference), which is of record in the application. The present reissue application was filed because I informed the assignee in 2003 that I believed the May reference precisely describes the peak reduction concept of the present application. Subsequent investigations and analyses on my part have only served to confirm that the May reference anticipates or renders obvious nearly all of the subject matter of the application, including all of the presently-claimed subject matter. I make this submission to explain how the claims pending on appeal are clearly identical to the teaching of May et. al.

The purpose of the remainder of this paper is to translate the text and equations of the May reference into a corresponding functional architecture for comparison with the architecture and claims of the current application. I will then explain the peak-reduction approach described in the present patent application, clarifying several terms used in the application to aid in comparing this approach to the May reference. Once the two approaches are clearly described using the same nomenclature, it is abundantly clear to anyone skilled in the field of peak-reduction processing that, at their broadest levels of disclosure and implementation, the May reference and the present application and claims are directed to the same peak-reduction concept; in fact, they are equivalent.

The discussion below follows a logical progression, with each section building on its predecessor. Section A sets forth the May et al. approach exactly as described in their paper. Section B converts that description, which uses analog signal processing notation, into the equivalent form using modern digital signal processing (DSP) notation, which simplifies subsequent analysis and comparison. May plainly described operations intended for digital implementation, but used a hybrid analog/digital notation which complicates detailed analysis. Converting to modern DSP notation facilitates detailed comparison to the present application. Section C translates the mathematical description of the May approach into its corresponding detailed functional architecture, a prerequisite for comparison with the functional architecture described in the present patent application. Section D sets forth the algorithm described in the present application, and identifies a critical deficiency in that description that would preclude anyone of skill in the art from being able to implement the presently-claimed subject matter. Section E provides this critical missing detail required to understand (and to implement) the presently-claimed subject matter, based on the detailed knowledge of an inventor, and clearly describes the equivalence between the presently-claimed subject matter and the subject matter described by the May reference. Finally, Section F explains the relevance of the multiple path versions of the architecture described in the present application.

A. May et al. Original Notation

The May et al. reference states a **precise definition of the set of signal sample times for which the described processing is to be applied.** ***"If the signal exceeds the amplitude threshold A_0 at the times t_n, then the corrected signal is"*** to be computed per the provided mathematical description. (Page 2475, col. 2). May et al. then provide two sets of equations precisely describing two alternative approaches for peak-reduction; ***each set of equations apply to the entire set of signal samples occurring at times $\{t_n\}$*****, not just the 'peak' samples.** The first approach, referred to as the "multiplicative correcting function" approach, computes a correcting function, k(t), then attenuates signal excursions above the threshold by multiplying the input signal by the corrective function. The second approach, referred to as the "additive correcting function" approach, computes a correcting function, k(t), then attenuates signal peaks by adding the input signal and the corrective function. The "additive correcting function" approach of May is most pertinent to the approach described in the present application.

May's 'additive' approach is precisely prescribed by three equations defining operations to be applied to the set of signal samples occurring at times $\{t_n\}$. These equations are duplicated below; each equation is provided an index to facilitate referencing in subsequent discussion. According to May, at each time instant $\{t_n\}$, where the nth time instant corresponds to the nth time the signal amplitude exceeds the defined threshold A_0, compute the (additive) corrected signal c(t) as:

$$c(t) = s(t) + k(t) \tag{1a}$$

$$k(t) = \sum_n A_n g(t - t_n) \tag{2a}$$

$$A_n = -\left(|s(t_n)| - A_0\right)\frac{s(t_n)}{|s(t_n)|} \tag{3a}$$

Equation (3a) computes a unique scale factor, $\mathbf{A_n}$, whose value depends only on the input signal value at time $\mathbf{t_n}$, and the amplitude threshold, $\mathbf{A_0}$. Equation (2a) forms a 'corrective function,' **k(t)**, as the sum of a scaled set of functions, **g(t)**, where each function is displaced in time, centered at time $\mathbf{t_n}$, and scaled by the factor $\mathbf{A_n}$. Equation (1a) forms the 'corrected signal' (i.e. peak-reduced) by adding the input signal to the corrective function, **k(t)**. These three equations, (1a -3a), precisely define the May et al. peak-reduction algorithm.

B. May et al. Digital Notation

May et al. used analog signal processing notation, yet nearly all modern peak-reduction processing is implemented using digital signal processing (**DSP**). While analog notation is often used by signal processing professionals educated prior to widespread adoption of a standard DSP notation, and clearly communicates the precise intent of the May reference, use of modern DSP notation in the discussion below facilitates one-to-one comparison of the description of May et al. with the present application and claims.. I have made the usual notational change from '**s(t)**' to '**s(m)**' to represent the **mth sample of the signal s(t).** Also, whereas the May reference defines times $\{\mathbf{t_n}\}$ by stating that **"the signal exceeds that amplitude threshold A_0 at times t_n,"** I use '**s($\mathbf{m_n}$)**' to represent signal samples exceeding the threshold at time t_n. The index set $\{\mathbf{m_n}\}$ identifies each time that a signal sample exceeds the threshold, with the secondary index

eliminating any notational ambiguity. For example, m_k is the index of the k^{th} sample whose amplitude exceeds the threshold; if the third signal sample, s(3), is the first whose amplitude exceeds the threshold, then $m_1=3$, etc.

With this notation, May's three key equations may be recast in the corresponding digital form:

$$c(m) = s(m) + k(m) \tag{1b}$$

$$k(m) = \sum_{m_n} A_{m_n} g(m - m_n) \;\; \forall m_n \ni n = 1,2,\cdots \tag{2b}$$

$$A_{m_n} = -\left(\left|s(m_n)\right| - A_0\right)\frac{s(m_n)}{\left|s(m_n)\right|} \tag{3b}$$

Equation (2b) uses standard mathematical notations '$\forall \mathbf{m_n}$' to indicate 'for all values of m_n', and '$\ni$ n = 1,2,···' to indicate 'such that n takes on all positive integer values'. Its meaning is then, "center a scaled version of g(m) at every instant for which the input signal amplitude exceeds the threshold." Equations 1b – 3b using digital notation are equivalent to May's equations labeled 1a-3a herein using analog notation.

C. <u>May et al. Functional Architecture</u>

This section converts May's equations directly into a corresponding detailed functional architecture. The most contentious issue in this translation concerns the issue of delaying the input signal. Specifically, is it possible to implement the May approach without delaying the input signal prior to adding the corrective function? This issue is easily resolved in unambiguous manner by noting the nature of a typical 'auxiliary function' as described in the May reference. On page 2475, column 2, May et al. describe the Gaussian function as an 'auxiliary function' suitable for their "multiplicative correcting function" approach. On page 2476 of their paper [in their equations (3-5)], they define the sinc function as an 'auxiliary function' suitable for their "additive correcting function" approach, one that is optimal for OFDM signals, as,

$$g(t) \approx \sin c(\pi Bt)e^{j\pi Bt}. \tag{4}$$

The critical point to appreciate is that the May et al. **auxiliary functions exhibit flat amplitude at time t=0**, and extend both backwards and forward about time t=0 for some finite period, or 'span'. Not only does May explicitly describe the auxiliary functions in this manner, it is obvious to anyone skilled in the signal processing art that both the auxiliary function and its derivatives must be continuous at t=0. This 'smoothness' condition assures that the spectrum of an auxiliary function is compact, which minimizes the magnitude of out-of-band spectral energy. Note that it precludes 'auxiliary functions' which abruptly jump from zero to unity at time zero; auxiliary functions described by May **must extend significantly to the left of the time origin.**

Now consider the processing prescribed by May et al. for any signal sample exceeding the amplitude threshold; consider for example the sample at t_7 in Figure 1. May's equation (1a) prescribes that the scaled auxiliary function (the sinc function defined above), centered at time instant t_7, must be subtracted from the input signal. **Unless the input signal samples are delayed by at least the span of the auxiliary (sinc) function to the left of the time origin, it is impossible to subtract all auxiliary function samples from the input signal,** since all input

signal samples occurring prior to t_7 are no longer available. It cannot be known that any sample will exceed the threshold until it actually occurs, yet May describes processing on both the signal and auxiliary function samples occurring prior to this sample (i.e. samples corresponding to the 'negative-time' samples of the auxiliary function). Consequently, it is ***impossible to implement the May reference teaching unless the input signal undergoes a fixed delay of at least the 'negative' time span of the auxiliary function.*** It is abundantly clear that this required **delay is fixed, rather than variable.**

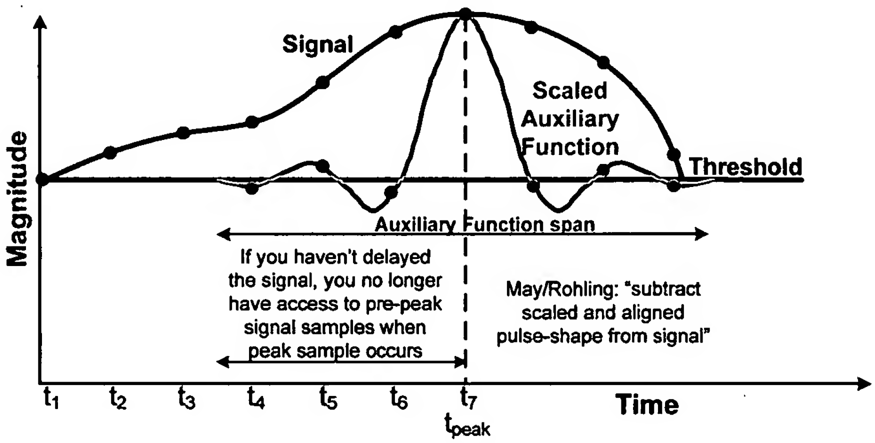

Figure 1.

Need to Delay the Input Signal by Half the Auxiliary Function Span

The assignee has made no attempt to explain how the May teaching could be implemented without delaying the input signal as I have described, notwithstanding that I have explained the delay requirement previously. It is quite simply impossible to implement the teaching of the May reference in any other way. Thus the assignee's statement in its Reply Brief that I "have not stated that May would work with a fixed delay" (at p. 2) is yet another misrepresentation of what I have said and is clearly wrong. The Examiner is correct: May requires a fixed delay.

Moreover, when I initially assigned my most junior engineer (circa mid-1998) the task of generating a detailed computer simulation of the approach of the present application, I made no mention of the inherent need to delay the signal, yet he immediately recognized that delay was needed. The need for delay on the signal path is obvious to anyone of ordinary skill in the art who doesn't have an interest in denying it. The assignee's statement in its Appeal Brief that I "did not assert that one of ordinary skill in the art would read May [to inherently require a delay]" (at p. 9) is ridiculous. The assignee's further argument that I understood the need for delay because of my alleged "intimacy with the issue" (Appeal Brief at p. 13) is equally ridiculous in light of the fact that I did not even tell my junior engineer to include a delay; he intuitively knew to do so. My prior submissions and my submission herein clearly explain the inherent requirement for delay in the May reference approach – there is no other way for a person of ordinary skill to view this issue. The assignee's statement that "more than ordinary creativity" (Appeal Brief at p. 9) would be needed to understand the need for delay in May is simply wrong.

Moreover, it is clear from the prior art that a delay was inherently required in the approach of May et al. For example, U.S. Patent No. 5,410,750 teaches to delay an input signal

by at least the fixed delay time required to generate an estimated interference canceling signal. (Col. 8, lines 14-28). Likewise, in the May et al. approach, the signal must also be delayed by the time required to generate the estimated peak canceling signal, which must be at least equal to half the auxiliary function span. Further examples include U.S. Patent Nos. 4,577,330 and 5,383,224, which teach to delay an input signal by at least the fixed delay time required to generate an estimated cross-polarization interference canceling signal. The '224 patent, for example, states:

> The delay circuit 16 delays the main polarization side digital signal string entered for a specified time. The delay circuit 16 is connected to a digital adder 17 and transmits the main polarization side digital signal string to the adder 17 after an appropriate time delay in order to compensate the delay caused by the cross polarization interference cancellation means 18. (Col. 3, lines 48-54; Fig. 1).

(See also '330 patent, Col. 5, line 54 – Col. 6, line 40; Fig. 1). Likewise, in the May et al. approach, the signal must also be delayed by the time required to generate the estimated peak canceling signal, which must be at least equal to half the auxiliary function span. In each of the situations described by these referenced prior art patents, ***when it is desired to cancel some unwanted attribute of a signal***, it is ***always necessary to delay that signal for as long as required to generate the 'corrective signal'*** which is to be subtracted from the signal to achieve the intended cancellation. In the May et al. reference, it was desired to cancel portions of a signal where the magnitude exceeded some threshold. As evidenced by, inter alia, these prior patents, a person of skill in the art would have known to use delay in the situation May et al. addressed and that delay was inherently required, i.e., these prior references clearly show that a person of skill in the art would know to **always** use delay in such a situation.

Having determined that fixed signal delay is needed in the May functional architecture, we may immediately associate four additional functional sub-blocks with May's definition of the set $\{t_n\}$ and equations (1b) – (3b): 1) an 'amplitude-based sample gate' to pass samples only at times, $\{t_n\}$, when the signal amplitude exceeds the threshold; 2) a 'sample scaling' to apply the scaling of equation (3b); 3) an 'auxiliary function placement' to generate the 'corrective function, **k(m)**, described in equation (2b); and, 4) a combiner, adding the delayed original samples to the corresponding corrective signal samples, as required by equation (1b).

At this point, the May et al. functional architecture may be accurately depicted. Ancillary functions of the initial (optional) band-limiting filter (shown in dashed borders) and the final linearizer will be discussed next, but the core of May's teaching is clearly depicted (at a top level) in Figure 2.

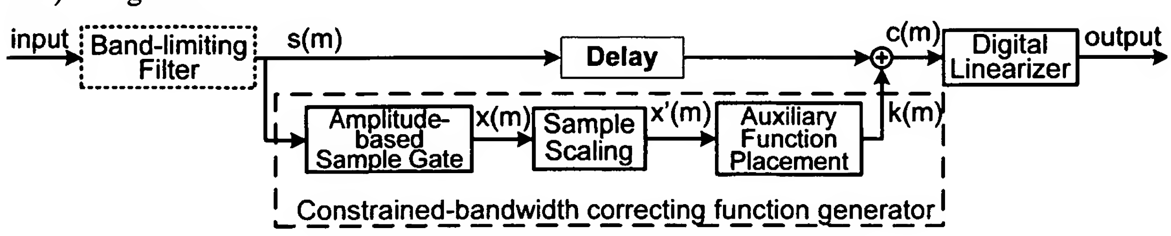

Figure 2. May et al. Functional Architecture

We have so far derived the key elements in the functional architecture depicted in Figure 2 directly from the description provided by May's teaching. The next few paragraphs explain the presence of the 'band-limiting filter' and the 'digital linearizer' operations in Figure 2.

1. Initial Band-Limiting Filter

May et al. do not explicitly describe the band-limiting filter responsible for generating the undesirable peaks, but its existence is implicit in those very peaks. Every data transmitter starts with a set of data n-tuples corresponding to some defined modulation 'constellation' of permissible data sets. The present application describes the passage of such data values thorough a 'pulse-spreading' filter. While this notation is accurate and standard, 'band-limiting' filter is equally accurate and standard. These two equally common descriptions reflect the fact that, if the impulse response width of any linear filter is increased by a factor, α, the bandwidth of that filter is simultaneously reduced by that same factor. This relationship between impulse response duration and filter bandwidth is well-known[1]. I am intentionally using the alternative notation in describing May's filters as a reminder that our comparison must consider them as different until later discussion establishes their relationship. It is clear that the band-limiting filter must be designed to ensure that the peak-reduced waveform spectral energy is sufficiently constrained to satisfy relevant regulatory spectral masks, such as those imposed by FCC 101.111, which regulates transmission signal spectra.

2. Digital Linearization

May et al. began their paper by noting that virtually all modern transmission systems have begun using some form of predistortion or linearization in order to achieve more linear amplification, and that therefore they would assume use of linearization/predistortion. Consequently, the linearizer function is included in the functional architecture depicted in Figure 2. Of course, any amplifier used in a communication transmitter must be 'substantially linear', since substantial linearity is a prerequisite for useful amplification. However, hundreds of technical papers and patents address the important need to increase the 'linear region' of any amplifier used with modern complex (often multi-channel) communication signals, and there is **obvious benefit in increasing the amplifier's 'linear region'**. Any amplifier exhibits a maximum output signal power level; **at the maximum level** of power, the transfer characteristic **must have a zero slope** (*from basic calculus*). Therefore, the only purely 'linear' amplifier must generate zero output power, since any other amplifier must have a slope change. ***The amplifier with the largest possible linear range (i.e. the 'most linear' amplifier) has a transfer characteristic identical to that depicted in May's Figure 1, reproduced below as our Figure 3 herein.*** Note that this 'most linear' amplifier characteristic can only be approached - never achieved - by cascading a linearizer and a 'substantially linear' amplifier.

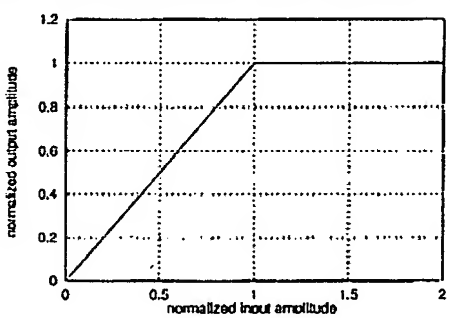

Figure 3.

Ideal limiter with normalized input and output amplitude, maximal input amplitude $A_0 = 1$.

Since digital linearization is so pervasive in modern systems, and since the focus of the May et al. paper was peak-reduction, rather than linearization, May et al. reasonably assumed

[1] E. Oran Brigham,. "Frequency Scaling." In ***The Fast Fourier Transform,*** Prentice-Hall, pp. 35-36, 1974.

their peak-reduction operated in cascade with an *ideal* linearizer, rather than a specific 'practical linearizer' exhibiting some specific degree of imperfection. The transfer characteristic of the cascade of a linearizer and any 'substantially linear' amplifier will more closely resemble the characteristic in Figure 3 than the original 'substantially linear' amplifier without linearization. Consequently, we may consider the transfer characteristic in Figure 3 as an ideal approximation of the composite transfer characteristic of any practical linearizer with any practical amplifier.

The assignee attempts to refute the Examiner's conclusions regarding May's teaching of linear amplification by attacking my credibility (Appeal Brief p. 13). My arguments are based on technical analysis. I leave to the PTO the determination as to whether a coherent technical analysis can be obviated by specious credibility attacks.

3. Detailed Descriptions of Functional Sub-Blocks

At this point in the development, we have derived the top-level functional architecture which exactly corresponds to May's teaching. However, to facilitate a detailed comparison with the current application, it is now necessary to expand the description of each of the functional blocks in figure 2 to **explicitly define the relationship between input and output samples for every critical functional sub-block** in order to complete the translation of May's hybrid textual/mathematical description into a corresponding detailed functional architecture. The fully translated functional architecture is shown in Figure 3 below.

a. Amplitude-Based Sample Gate

The functional sub-block labeled 'amplitude-based sample gate' in Figure 3 simply passes intact any input signal sample, **s(n)**, whose amplitude exceeds the amplitude threshold $\boldsymbol{A_0}$, and resets to a zero value any input signal sample whose amplitude fails to exceed that threshold. Since the subsequent functional sub-blocks take no action in response to samples having a value of zero, this accomplishes exactly the prescription of May to act **only** on samples having amplitude greater than the threshold. Thus,

$$x(m)=\begin{cases} 0 & if \ |s(m)|\le A_0 \\ s(m) & otherwise \end{cases} \tag{5}$$

b. Sample Scaling

May et al. describes the ***generation of a specific scale factor in response to each sample whose amplitude exceeds the threshold.*** They define that scale factor as

$$A_{m_n}=-\left(|s(m_n)|-A_0\right)\frac{s(m_n)}{|s(m_n)|} \tag{6}$$

The "sample scaling" functional sub-block of Figure 3 simply **replaces each non-zero input sample with its corresponding scale factor.** Thus,

$$x'(m)=\begin{cases} (0) & if \ |s(m)|\le A_0 \\ \left[-\left(|s(m)|-A_0\right)\frac{s(m)}{|s(m)|}\right] & if \ |s(m)|> A_0 \end{cases} \tag{7}$$

c. Linear Filter Interpretation

At this point, it is instructive to consider what the output of a linear filter would be if x′(m) were its input? From linear theory we know that we can represent the output, **z(m)**, of any linear finite-impulse response (FIR) filter of length 2K+1, as the convolution of the function that describes the filter input, **x′(m)**, with that filter's impulse response function, **h(m)**. We begin by converting the description of individual sample values of the filter input sequence, **x′(m)**, into the actual equation for the corresponding function,

$$\mathbf{x}'(\mathbf{m}) = \sum_{m_n} - \left[\left(|s(m)| - A_0\right) \frac{s(m)}{|s(m)|} \right] \delta(m - m_n) \forall m_n \ni n = 1,2,\cdots \tag{8}$$

The so-called 'delta' notation is used to convert the set of individual samples into a functional description, since these values are not simply scalars – they represent a specific scalar value occurring at a specific instant. The delta notation, familiar to those skilled in the signal processing art, describes this particular attribute. Even though anyone skilled in signal processing would recognize this, in translating May's textual/mathematical description into the corresponding functional architecture, care is taken to preserve the exact May teaching. This mathematical rigor will clearly show how May's teaching precisely translates into a specific corresponding functional architecture.

The **filter impulse response, h(m),** consists of the sequential values {h(-K), h(-K-1), . . . , h(0), . . . , h(K-1), h(K)}, so the filter output may be written as the familiar convolution

$$z(m) \equiv \sum_k h(k)\mathbf{x}'(m-k) = \sum_k h(k) \left[\sum_{m_n} A_{m_n} \delta(m - m_n - k)\_\forall m_n \ni n = 1,2,\cdots \right], \tag{9}$$

This is immediately recognized, by any person of ordinary skill in the art[2], as

$$z(m) = \sum_{m_n} A_{m_n} \mathbf{h}(\mathbf{m} - \mathbf{m}_n) \; \forall m_n \ni n = 1,2,\cdots, \tag{10}$$

The key feature to note: equation (10) ***centers*** a **scaled replica of the filter impulse response at every sample where the input signal exceeds the threshold.** The implication of this fact is obvious. If the scaled impulses defined by equation (7) are processed through any **linear filter whose impulse response function is equal to May's 'auxiliary function,'** the output exactly equals that prescribed by May. Consequently, we may evolve the top-level functional architecture of figure 2 into the more detailed functional architecture depicted in Figure 4.

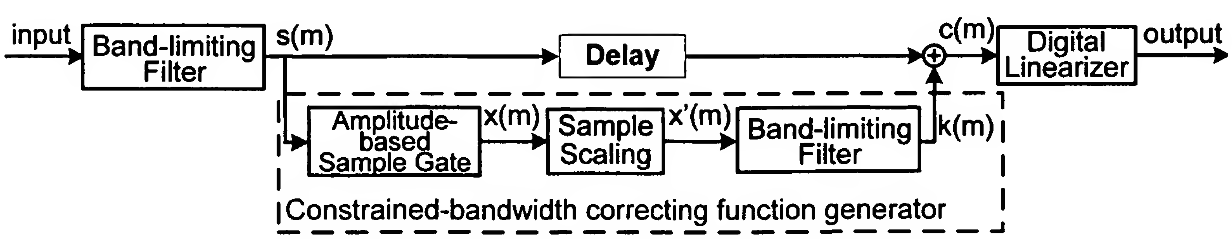

Figure 4. Equivalent May et al. Functional Architecture

[2] Bracewell, R. "The Sifting Property." In ***The Fourier Transform and Its Applications, 3rd ed.***, New York: McGraw-Hill, pp. 74-77, 1999.

The corresponding description of May's corrective sequence, **k(m)**, matches equation (2b) as

$$k(m)=\sum_{n} A_{m_n} g(m-m_n)\_\forall m_n \ni n=1,2,\cdots. \tag{11}$$

Note that the same label is given to both the initial band-limiting filter and the band-limiting filter that follows the sample scaling operation. These filters must be very nearly identical in order to ensure compliance with regulatory spectral masks. The technical proof is straightforward: the power spectral density of the sample streams, **x(m)** and **x'(m)** will be relatively flat, since the small set of samples in each group of samples exceeding the threshold appear similar to a brief 'impulse' – and the spectrum of any impulse is a flat spectrum. While this heuristic reasoning is only approximately true, the actual spectrum is nearly as flat as the sinc-function spectrum associated with random data samples which constitute the input to the first band-limiting filter. Therefore, the attenuation characteristic of both band-limiting filters, which are defined by the need to meet identical spectral mask constraints at the combiner output, must be very nearly identical. For purposes of the present discussion, it is assumed that the two filters are identical, with each filter impulse response equal to May's 'auxiliary function'.

Obviously the functional architecture of the May approach depicted in Figure 4 bears close similarity to the approach of the present application. However, in order to precisely compare the two approaches, we must translate the processing described in the current patent application into an equally well-defined detailed functional architecture. This is accomplished in the next section.

D. Functional Architecture of the Present Application

The functional architecture of the peak-reduction approach of the present application is depicted in Figure 5. Note that I have depicted only a **single correction path**, as opposed to the "two or more" such paths described in the present application. This will greatly simplify subsequent discussion, and is directly relevant, since the present application **claims** clearly cover a single path, as well as the "two or more" paths described in the actual patent description. Once the relationship between this single-path version and May et al. is understood, we can easily extend the comparison to any number of paths; this extension is described in Section F below.

Capital letters are used to distinguish potentially distinct signals in Figures 4 and 5, since it *cannot yet be assumed that they are identical.* Several further issues are dealt with in appropriate detail in subsequent paragraphs. Initially, issues regarding the **filter and linearizer operations** permit immediate resolution.

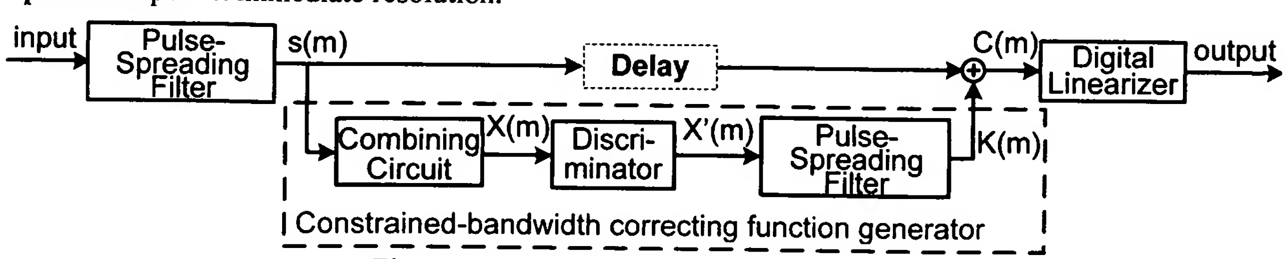

Figure 5. May et al. Functional Architecture

1. Pulse-Spreading vs. Band-limiting Filter Comparison

It is readily apparent that, labels notwithstanding, May et al.'s 'band-limiting filter' (aka 'auxiliary function') and the 'pulse-spreading filter' of the present application are equivalent. Pulse-spreading (i.e. increasing the time-domain pulse width) proportionally reduces its spectral bandwidth, thereby 'band-limiting' the signal[3]. This qualitative observation can be demonstrated by a quantitative example. Figure 6 compares May's sinc 'auxiliary function' with the Root-Nyquist (α=0.22) filter impulse response described by the present application. Since the primary peak-reduction properties of both approaches primarily depend on the main-lobes and zero-crossings of these functions, and inspection reveals these to be indistinguishable, it is obvious that these two functional sub-blocks (i.e. 'pulse-spreading filter' and 'band-limiting filter') are very nearly identical. Succinctly, the nearly-identical filters used in both approaches are clearly equivalent, and their functional sub-blocks in Figures 4 and 5 are interchangeable.

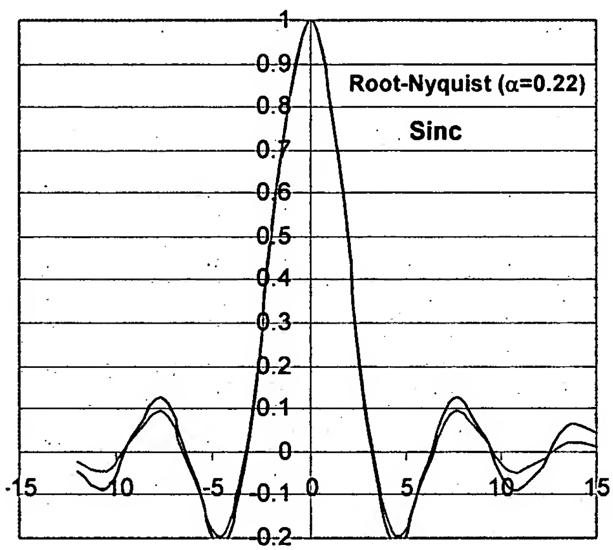

Figure 6.

Root-Nyquist (α=0.22) Pulse vs. Sinc Function: Time-Domain Waveforms

2. Linearization Comparison

After carefully scrutinizing the present assignee's arguments regarding the use of linearization, I still do not understand them. Both patents in the present family (6,104,761, 6,366,619) clearly included digital linearization, and the May reference explicitly assumed the use of digital linearization. The assignee's arguments, which have seemingly been accepted by the Examiner, nonetheless cite an important distinction between these two cascades of peak-reduction and linearization. As best as I can infer from the arguments, the claimed distinction is that May et al. assumed an **ideal** linearizer characteristic, whereas the present application assumes an imperfect linearizer. However, it is important to recall that the cascade of an ideal linearizer with a substantially linear amplifier yields the ideal linear amplifier transfer characteristic depicted in Figure 7, which is identical to Figure 1 of the May reference.

[3] Ibid. See "Scaling property."

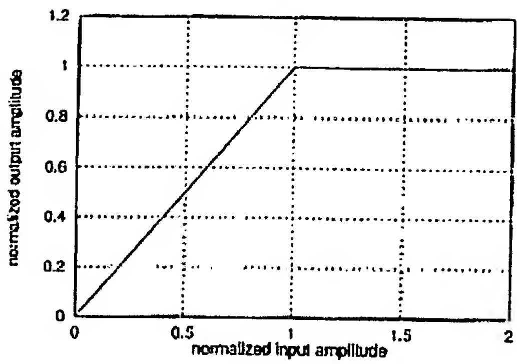

Figure 7.

Ideal limiter with normalized input and output amplitude, maximal input amplitude $A_0 = 1$.

The assignee's U.S. Patent 6,104,761 clearly addresses this critical issue:

> Within substantially linear amplifier 146, digital linearizer 148 alters constrained-envelope signal stream 144 into a pre-distorted digital signal stream 154. Pre-distorted digital signal stream 154 is made non-linear ***in just the right manner to compensate for non-linearities*** within digital-to-analog converter 150 and RF amplifying circuit 152, hence linearizing substantially linear amplifier 146. (Col. 12, lines 29-36).

It is clear that these words from the '761 patent precisely describe the usual **goal of linearization**, i.e., use predistortion to compensate for residual nonlinearity in a 'substantially linear amplifier' so that the transfer characteristic of the **cascade of the linearizer and substantially linear amplifier is far more linear than the 'substantially linear amplifier' by itself**. Those skilled in the art of linearization prior to this patent were well aware that the cascade of linearizer and amplifier is intended to produce an equivalent transfer function exactly like that in Figure 7. For example, Andreoli clearly described the goal of digital linearization for an RF high power amplifier (**HPA**):

> A predistorting device designed to counteract the nonlinear distortions introduced by the amplifier has to implement two nonlinear functions H and Φ that globally invert G and Ψ. The **cascade of the predistorter with the HPA** gives rise to a pseudo-linear device (i.e. **an ideal soft-limiting device**) as showed in figure 2, where the resulting AM/AM and AM/PM curves for the predistorted amplifier are shown.[4]

Andreoli's Figure 2 is reproduced below as our Figure 8.

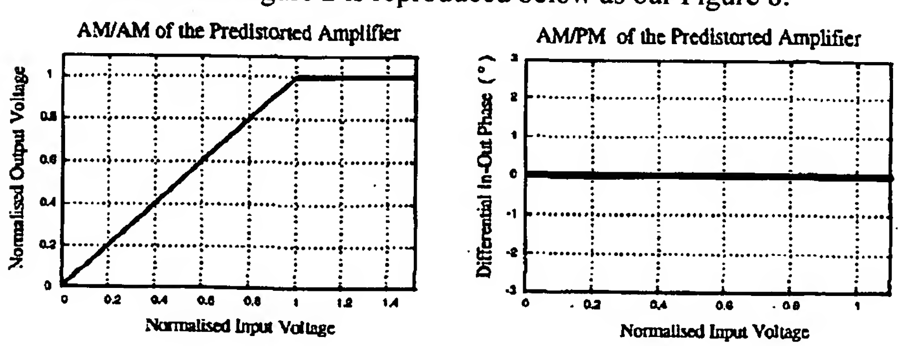

Figure 8.

Predistorted Amplifier Distortion Curves

[4] S. Andreoli, et. al., "Digital Linearizer for RF Amplifiers," IEEE Transactions on Broadcasting, vol. 43, March 1997, pp. 12-19.

May et al.'s predistorted/linearized AM/AM curve is identical to Andreoli's "ideal soft-limiting device". This is a clear indication that those skilled in the art of linearization agreed on the goal of linearization well before the filing date of the present application. The facts simply cannot be reconciled with the assignee's arguments regarding linearization.

I disagree with the assignee and its expert, Mr. Birch, with respect to Mr. Birch's arguments regarding amplifier "linearity". Mr. Birch stated in a declaration: "I've reviewed the teaching of Dent patent 5262734. Dent describes a linear RF power amplifier 10 that produces intermodulation products. . ." Mr. Birch's statement is erroneous because the Dent patent uses improper terminology; a <u>LINEAR</u> amplifier cannot generate intermodulation products. "Linearity" actually means something very specific in engineering: the output equals a scalar times the input. Hence a LINEAR amplifier can never generate nonlinear intermodulation products. The fact that a patent has been issued that used sloppy terminology should have no bearing on the present application. Engineering terms such as "linearity" should be used in the manner that engineering science has always taught.

In fact, there is no such thing in the real world as a LINEAR amplifier; all real amplifiers exhibit some maximum output power at which the transfer characteristic slope must equal zero. Consequently, since linearity implies a constant slope everywhere, the only truly LINEAR amplifier would have zero output, which of course is meaningless. Engineers skilled in linearization refer to an amplifier as being "**more linear**" if it has **a larger range over which the transfer characteristic is absolutely linear**. A cursory examination of the AM/AM transfer characteristic in Figure 8 clearly shows that this curve maximizes the linear range possible for this specific maximum output voltage. Hence the common description of this characteristic as the "most linear" or "ideally linearized" amplifier, its change in slope notwithstanding.

Based on the assignee's argument regarding linearization, it is apparently critical that a linearizer designer ensure that the design either is **very poor** or is **nearly perfect** to avoid infringement of the pending claims of the present application. If the linearizer designer is incompetent, the cascade of the linearizer with the amplifier will not yield a composite "substantially linear amplifier." If the designer is successful in designing a near-ideal linearizer, such that the cascaded linearizer-amplifier characteristic closely approximates Figure 8, presumably such design would also not infringe claims of the present application, since this performance was described in the May reference prior to the filing date of the present application. The presently pending claims apparently cover all linearizers of mediocre (i.e. not too poor, and not too good) design quality. Since I am irrevocably led to this conclusion by the assignee's arguments in this application, but am baffled by the logic, I will simply include the 'digital linearizer' in both functional architectures for purposes of analysis.

This leaves a **single remaining point of potential discrepancy between the functional architectures of May et al. and the present application**. This potential discrepancy is depicted in Figures 4 and 5, respectively: as the **first two sub-blocks on the non-delayed signal path.** We already have a precise description and a detailed functional architecture description of the May et al. approach; we need a similarly precise detailed functional architecture embodying the present application. To develop this detailed functional architecture, we will first examine the description provided in the patent application. Since this description will be shown to be far too vague to permit one skilled in the art to develop the detailed functional architecture we need for comparison with May, I then provide a more detailed description to resolve a key ambiguity left by the patent description.

U.S. Patent No. 6,104,761, col. 11, lines 1-14, describes using a 'constrained-envelope generator' **106** to combine a threshold signal **120** with a signal stream **84** to produce a difference signal stream **124**. This difference signal stream is passed to the input of a discriminator **128** to produce an error signal stream **130**. This error stream is then passed to the input of a pulse-spreading filter **134** which produces the constrained bandwidth error signal stream **108**. Pulse-spreading filter **134** is substantially identical to the first pulse-spreading filter in Figure 4 detailing the May et al. functional architecture. *Note that, since 'on-time' and 'off-time' processing is identical, I have deleted these extraneous qualifiers to increase clarity.*

The actual **'combining circuit' processing** is described by the '761 patent as follows:

> Threshold signal **120** and signal stream **86** are combined in a complex summing or combining circuit **122** to produce a difference signal stream **124**. Difference signal stream **124** is made up of a series of difference pulses **126** whose values are the difference between the values of equivalent pulses **82** and the value of threshold signal **120**. Since any given pulse 82 may have a value greater than, equal to, or less than the value of threshold signal **120**, difference signal stream **124** would normally be made up of a combination of difference pulses **126** having positive, zero, and negative values. (Col 9, lines 25-37; as before, the qualifiers 'off-time' and 'on-time' are deleted).

In layman's terms, the **combining circuit simply replaces all input samples with the difference between that sample and the threshold.**

The actual **'discriminator' processing'** is described by the '761 patent as follows:

> Difference signal stream **124** is passed to the input of a discriminator **128** to produce an error signal stream **130**. In the preferred embodiment, error signal stream **130** is a variation of difference signal stream **124** in which all difference pulses **126** having positive values are passed unchanged as error pulses **132** while all other difference pulses **126** are passed as zero-value pulses (i.e. eliminated). (Col 9, lines 38-45; I again delete the extraneous qualifiers).

In layman's terms, the **discriminator simply replaces all non-positive samples with zero-valued samples.**

Clearly, the approach described by the present application generates a series of samples/pulses which have zero value for all input signal samples whose amplitude doesn't exceed the threshold, while replacing all signal samples whose amplitude does exceed the threshold by the "complex summing" of the signal sample and the threshold. This sequence of samples constitutes the input to their 'pulse-spreading' filter, the output of which is combined with the appropriately delayed original input signal.

While this description of the 'combining circuit' and 'discriminator' appears simple, a person of ordinary skill in the art would not implement the 'complex combiner' from the description provided. Following the patent, a person of ordinary skill would logically convert the real threshold value into its corresponding complex value, with a real part equal to the threshold and an imaginary part equal to zero. That person would then compute the 'complex difference' between these two values as a complex value with a real part equal to the difference between the real part of the signal sample and the threshold, and an imaginary part equal to the imaginary part of the signal. However, this would produce a worthless result, because the

'complex combiner' operates in a much different manner than one of ordinary skill would conclude based on the description provided in the patent. The purpose of this submission is to provide a very clear and understandable comparison between the May reference and the present application To provide such a comparison, it is necessary to provide a much clearer description of 'complex difference' than is provided by the present application, since it is critical to proper operation of this invention.

E. Technical Description of the Present Application

Figure 9 depicts the "inventors' version" of the functional architecture of the approach of the present application. The 'complex combiner,' (with its poorly defined 'complex difference' operation) and the discriminator have been replaced by an 'excursion generator.' A clear description of the operation of the excursion generator will eliminate all ambiguity regarding processing pursuant to the approach of the present application, which will enable an accurate comparison between it and the May reference.

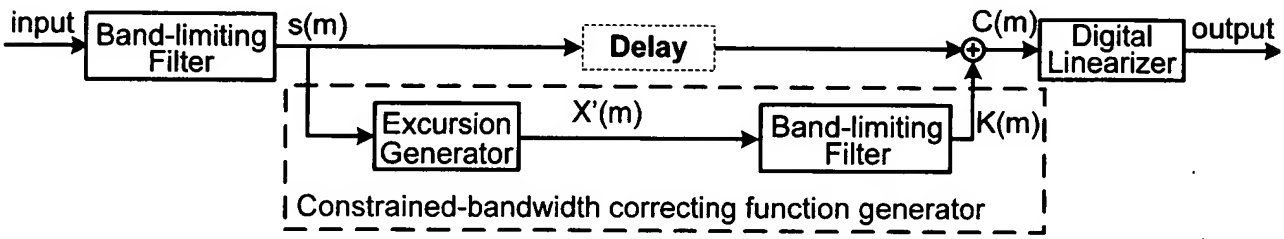

Figure 9. Inventors' Version of CERN Functional Architecture

To understand the function of an 'excursion generator', it is first necessary to understand the operation of clipping. A 'clipper' is a nonlinear operator defined by equation (12), which reproduces its real input signal only when the signal amplitude is less than some defined real threshold, A_0.

$$Clipper_{out}(m) \equiv \begin{cases} s(m)\_if\_|s(m)| \le A_0 \\ A_0\_if\_|s(m)| > A_0 \end{cases} \tag{12}$$

The complex equivalent form of the clipping operator is defined in equation (13)

$$Clipper_{out}(m) \equiv \begin{cases} s(m)\_if\_|s(m)| \le A_0 \\ A_0\left[\dfrac{s(m)}{|s(m)|}\right]\_if\_|s(m)| > A_0 \end{cases} \tag{13}$$

The signal '**excursion**' is simply defined as **that part of the input signal 'clipped away' by a clipper.**

$$X'(m) = \mathbf{Excursion}(m) \equiv \begin{cases} 0\_if\_|s(m)| \le A_0 \\ s(m) - A_0\left[\dfrac{s(m)}{|s(m)|}\right]\_if\_|s(m)| > A_0 \end{cases} \tag{14}$$

These precise equations permit a simple heuristic interpretation. Figure 10 depicts an arbitrary signal, an amplitude threshold, and the corresponding signal 'excursion' waveform in

graphical form. The signal 10 is a complex function of time. Since it is complex, each sample of the signal can be represented as a pair of values, an (I,Q) pair of in-phase and quadrature-phase values which completely define its instantaneous magnitude and phase. A clipper replaces all those signal 'excursions' (30 and 40) beyond the defined amplitude threshold 20 by signal segments (50 and 60) having the same phase as the input signal, but with magnitude equal to the amplitude threshold value. An 'excursion generator' output consists *only* of all excursions associated with the input signal and a specified amplitude threshold.

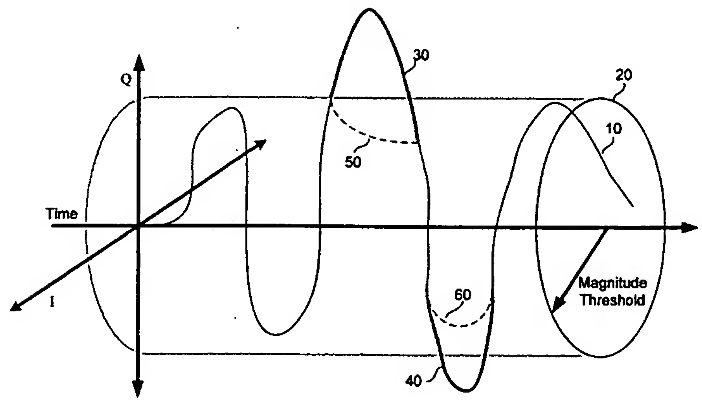

Figure 10.

Graphical Description of a Signal, an Amplitude Threshold, and the Corresponding 'Excursion'

The set of signal sample instants, $\mathbf{m_n}$, and corresponding samples, $\mathbf{s(m_n)}$, to be scaled in the May et al. processing are identical to the set of time instants and samples produced by our excursion generator as defined. May et al. define their 'excursion' as follows: "If the signal exceeds the amplitude threshold $\mathbf{A_0}$ at the times $\mathbf{t_n}$, then the corrected signal $\mathbf{c(t)}$ is . . ." and their algebraic description of the correction and corrected signals immediately follows. The **key point** is that ***the set of samples May et al. describe as being scaled is identical to that set of samples appearing at the output of our excursion generator.***

The present application describes the excursion generation operation by the vague terms 'combining circuit' and 'discriminator,' apparently in an attempt to obtain broader claim scope. The rationale appearing to be that the 'combining circuit' combines the input signal samples and the amplitude threshold to produce difference samples, and the discriminator zeroes out all but positive difference samples.

In the May approach, all signal samples with amplitude less than or equal to the threshold are set to zero, and then all non-zero output samples are replaced by the 'scale factor' computed according to equation (6). Finally, the stream of 'scale factors' forms the input to the band-limiting filter.

In the approach of the present application, the 'complex difference' between all signal sample values and the threshold are passed to the discriminator function, which simply sets to zero all but the positive samples. This 'excursion' sample stream forms the input to the pulse-spreading filter.

It is apparent that the two approaches are performing very similar processing and, as proven below, the two approaches are in fact identical for the case of a single processing path.

The May et al. filter input sequence values are described by:

$$x'(m)=\begin{cases}(0)\_if\_|s(m)|\le A_0\\ \left[-(|s(m)|-A_0)\frac{s(m)}{|s(m)|}\right]\_if\_|s(m)|>A_0\end{cases} \tag{15}$$

The present application describes the filter input sequence values by:

$$X'(m)=\textbf{Excursion}(m)\equiv\begin{cases}0\_if\_|s(m)|\le A_0\\ s(m)-A_0\left[\frac{s(m)}{|s(m)|}\right]\_if\_|s(m)|>A_0\end{cases} \tag{16}$$

Since,

$$-(|s(m)|-A_0)\frac{s(m)}{|s(m)|}=-\left[\frac{s(m)|s(m)|-A_0s(m)}{|s(m)|}\right]=-\left[s(m)-A_0\frac{s(m)}{|s(m)|}\right], \tag{17}$$

and since the **CERN implementation** ***subtracted*** **(hence the sign 'flip') filtered excursion values from the delayed signal**, the reason for my assertion to the assignee in August 2003 (8/13/03 email to Paul Bernkopf) that the May reference was "a precise description of CERN [the subject matter of the present application] - four months prior to our filing date" [the filing date of the parent of the present application] can be readily appreciated.

It is obvious that ***these two approaches are mathematically identical*** if the pulse-spreading filter impulse response and the auxiliary function are identical, **AND** ***if the implementation of the present application uses a single processing path***, as has been assumed in the development to this point. It is now possible to accurately resolve the contention between me and the assignee regarding the relationship between the present application and the May reference. **To the extent the present application encompasses the use of a single processing path, it is clearly identical to the teaching of the May reference.** However, since May et al. made no mention of using more than one processing path to generate their corrective signal, the present application would potentially ***be distinct from May if the claims are restricted to multiple corrective signal processing paths.***

F. Multiple Correction Path Implementation of the Present Application

In both the patent application and in our implementations, the approach of the present application demultiplexed the input signal sample stream into N sub-streams consisting of equally-spaced samples at a sample rate equal to N^{-1} times the input sample rate; with samples in each of the parallel sub-paths spaced by exactly the baud interval of the signal. The functional architecture of the multiple correction path implementation is depicted in Figure 11.

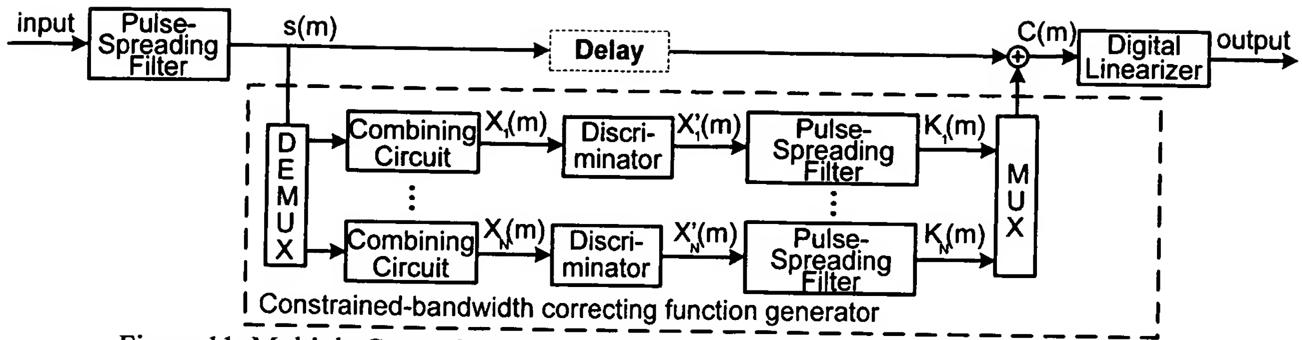

Figure 11. Multiple Correction Path Functional Architecture of the Present Application

The assignee's U.S. Patents 6,104,761 and 6,366,619 and their derivatives (one of which is the currently appealed application), are quite clear in teaching use of ***"at least two" parallel correction paths*, each of which processes samples spaced by baud intervals.** The '761 patent states:

> [P]ulse-spreading filter **76** produces <u>**at least two**</u> **(only two in the preferred embodiment)** output filtered signal pulses **78**, i.e. complex samples of filtered signal stream **74**, for each input phase-point pulse **66** received. This is demonstrated in FIG. 4 where filtered signal stream **74** possesses two filtered-signal pulses **78** per unit baud interval **64**. In the preferred embodiment, filtered-signal pulses **78** consist of alternating on-time pulses **80**, i.e. samples of filtered signal stream at integral unit baud intervals **64**, and off-time pulses **82**, i.e. samples of filtered signal stream **74** between integral unit baud intervals. In effect, filtered signal stream **74** is made up of two interleaved data streams, and on-time signal stream **84** and an off-time signal stream **86**. (Col. 6, lines 61-64; emphasis added).

The teaching of the '619 patent is clear that **each path consists of baud-spaced sample streams:**

> In FIG. 5, exemplary on-time filtered phase points **96** are located at integral-baud times (t_0, t_1, t_2, etc.), whereas exemplary off-time phase points **98** are located at fractional-baud (non-integral-baud) times ($t_{0.5}$, $t_{1.5}$, $t_{2.5}$, etc.). (Col. 7, lines 37-41).

Col 8, line 66 then begins the description of the processing applied to all off-time samples in order to generate the off-time pulse-spreading filter **134** output sample stream. The description continues in Col. 9:

> Off-time pulse-spreading filter **134** produces off-time constrained-bandwidth error signal stream **108** and completes the action of off-time constrained-envelope generator **106**. (Col. 9, lines 57-60).

The corresponding on-time pulse processing is described at Col 10, line 57 through Col 11, line 19.

While the description provided in the patent application under appeal clearly describes "at least two" paths, the actual claims on appeal fail to explicitly make this distinction, and thus cover even a single-path version of the concept. However, it should be quite clear from the discussion above that the single-path version of the approach of the present application is

substantially identical to the approach of May et al. My assertion to the assignee in August 2003 that the May reference provides "a precise description of CERN [the described and claimed subject matter of the present application] - four months prior to our filing date" is absolutely correct – ***if the claims fail to distinguish between using single-path and multiple-path processing to generate the 'constrained-bandwidth error streams'***. If the assignee continues to claim entitlement to the single-path teaching of May et al., I will be obligated under M.P.E.P. Section 2001.05, 'Materiality Under 37 C.F.R. 1.56(b)(2)' to continue refuting this misrepresentation. However, **if the claims were amended to clearly cover ONLY the use of two or more processing paths to generate the 'constrained-bandwidth error streams," each operating on baud-spaced samples, I would not have a continuing obligation to point out the applicability of the teachings of May et al. with respect to such claims.** To the extent any such multiple path claims may be considered obvious, I am not aware that the assignee has made any misrepresentation in this regard and thus I presently make no conjecture on that issue.

I further declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true, and further, that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code, and that such willful false statements may jeopardize the validity of the application or any patent issuing thereon.

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